

1 **METHOD OF OPTIMIZING COMPRESSION RATE IN ADAPTIVE**
2 **DIFFERENTIAL PULSE CODE MODULATION (ADPCM)**

3 **BACKGROUND OF THE INVENTION**

4 1. Field of the Invention

5 The present invention relates to a method of optimizing the compression
6 rate in Adaptive Differential Pulse Code Modulation (ADPCM), especially to a
7 modified pulse code modulation technique that employs a prognostic code
8 converter to generate variable length codes on top of the ADPCM coding, based
9 on the probability of occurrence of data bits, so as to further reduce the
10 compressed data size and to increase the compression rate of the conventional
11 ADPCM coding.

12 2. Description of Related Arts

13 The main difference between the pulse code modulation (PCM) and the
14 Adaptive Differential Pulse Code Modulation (ADPCM) lies in the compression
15 format, where the PCM coding is mainly used to convert analog signals to digital
16 signals in the audio signal processing using a fixed sampling frequency. This
17 PCM coding has the advantage of realistic sound reproduction, but a large
18 memory is required for such operation. ADPCM coding is an adaptive coding
19 technique that incorporates an adaptive predictor in the compression coding
20 process to produce compressed data with suitable compression rate. For example,
21 a 16-bit data sample can be compressed according to a 4:1 ratio to a 4-bit data
22 sample; and conversely, the compressed data can be decompressed according to a
23 1:4 ratio to restore it to the original signal waveform of a 16-bit string.

24 The application of ADPCM for compressing audio and speech signals is

1 aimed at easing the transmission bandwidth and reducing the memory storage
2 requirements. However, the main disadvantage of ADPCM lies in the low
3 compression rate. Although the memory requirements for the ADPCM codes are
4 fewer than those for the PCM codes, the compression rate is still not sufficient to
5 meet the memory requirements for portable electronic devices, for example CD
6 players.

7 From numerous observations, it is found that data contents in adjacent
8 data blocks are subject to less variation than non-adjacent data blocks. If the
9 locality of data in the compression coding can be analyzed, the compression rate
10 can be further improved, A modified pulse code modulation technique based on
11 Huffman coding can be employed by first creating a look-up table with variable
12 length codes that match the occurrence rate of data bits. This modified pulse code
13 modulation process, using variable-length or run-length coding, can be a
14 second-level coding process on top of the conventional ADPCM for achieving a
15 higher compression rate in the audio signal processing.

16 In general, both coding techniques have been used in digital signal
17 processing. For example, a portable CD player may be installed with an
18 electronic anti-shock system (EASS) to maintain constant quality of audio output
19 and to create a buffering effect against physical vibrations of the CD player. The
20 electronic anti-shock system (EASS) shown in Fig. 4 with a single output
21 channel has an ADPCM encoder (71) on the front end and an ADPCM decoder
22 (73) on the back-end. The ADPCM encoder (71) is used to generate ADPCM
23 codes from input data samples which are then saved in the random access
24 memory (72); and conversely, the compressed data retrieved from the random

1 access memory (72) are processed by the ADPCM decoder (73) for decoding the
2 ADPCM codes to the original waveform of audio signals en route to an audio
3 signal processor (74) for audio output.

4 The compression coding for the above-mentioned ADPCM encoder (71)
5 can be both a 4-bit mode and 3-bit mode. Since all current audio equipment has at
6 least two audio channels, the 4-bit operation mode is selected in this example for
7 calculation of the bit rate with a sampling frequency of 44.1 KHz:

8
$$4(\text{bits}) \times 44100 \times 2 (\text{number of audio channels}) = 352,800 \text{ Kbps}$$

9 If the operation is in 3-bit operation mode, the bit rate is:

10
$$3(\text{bits}) \times 44100 \times 2 (\text{number of audio channels}) = 264,600 \text{ Kbps}$$

11 If the memory capacity in the above anti-shock buffering system is 16M,
12 Bits, as in the present example, the required buffering time for the two operation
13 modes can be:

14 4-bit mode: $16,000,000 \div 352,800 = 45.35 \text{ (sec)}$

15 3-bit mode: $16,000,000 \div 264,600 = 60.46 \text{ (sec)}$

16 From the foregoing, it is apparent that the bit rate and the memory
17 capacity directly affects the buffering time of the EASS. Since the data saving
18 operation in the EASS has to be sustained for certain time period in order to
19 create an anti-shock buffering effect, and the buffering time is dependent on the
20 compression rate in the coding process and the capacity of random access
21 memory used, the performance of an ADPCM-based CD player embedded with
22 EASS can also be affected.

23 However, the main disadvantage of ADPCM lies in the low compression
24 rate, as the resulting compression rate of the conventional ADPCM is not enough

1 for efficient utilization of random access memory as in the case of the
2 ADPCM-based CD players. There are some more advanced compression
3 algorithms, such as the MPEG/layer I and II, which are able to produce
4 reasonably good sound quality using much less memory than ADPCM.

5 Under the precondition not to increase random access memory, the only
6 way to improve the performance of EASS for ADPCM-based CD players,
7 without incurring additional hardware costs, is to increase the bit rate in the
8 compression process. If the compression rate of ADPCM is increased, the
9 performance of EASS can be improved with lossless code modulation. Another
10 benefit of EASS is the power saving consideration for portable CD players as the
11 longer the CD servo can be suspended by the buffering effect, the less system
12 power is used.

13 SUMMARY OF THE INVENTION

14 The main object of the present invention is to provide a method of
15 optimizing the compression rate in Adaptive Differential Pulse Code Modulation
16 (ADPCM) that uses variable length coding based on the probability of
17 occurrence of data bits to further reduce the size of compressed data and improve
18 the performance over conventional adaptive differential pulse code modulation
19 (ADPCM).

20 To this end, the modified pulse code modulation technique employs a
21 prognostic code converter to generate variable length codes on top of the
22 adaptive differential pulse code modulation (ADPCM) coding. Variable length
23 codes are generated according to the probability of occurrence of the data bits. In
24 the variable length coding process, a bit string with the highest occurrence rate

1 will be assigned a variable length code with the shortest length, whereas the bit
2 string with the lowest occurrence rate will be assigned variable length code with
3 the longest length. Conversely, when decoding, the compressed data is first
4 processed by a variable length decoder to reconstruct the pure ADPCM coding
5 and then to the original audio signal in a lossless audio signal processing.

6 The features and structure of the present invention will be more clearly
7 understood when taken in conjunction with the accompanying figures.

8 BRIEF DESCRIPTION OF THE DRAWINGS

9 Fig. 1 is system architecture of the present invention;

10 Fig. 2 is an example of the probability table of the occurrence of bit
11 strings in a certain data block;

12 Fig. 3 is a table of the variable length codes matching the probability of
13 occurrence of data bits; and

14 Fig. 4 is a block diagram of the conventional electronic anti-shock
15 system (EASS).

16 DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

17 The present invention provides a method of optimizing the compression
18 rate in a adaptive differential pulse code modulation (ADPCM) by employing a
19 prognostic code converter with a system architecture as shown in Fig. 1,
20 comprising:

21 an ADPCM encoder (11) for encoding with an ADPCM coding
22 technique;

23 a random access memory (12) for temporary saving of output codes from
24 the ADPCM encoder (11) to provide a buffering effect;

1 an ADPCM decoder (13) for restoring the compressed ADPCM codes
2 retrieved from the random access memory (12) to the original audio signals;
3 a variable-length encoder (14), which is installed in between the
4 ADPCM encoder (11) and the random access memory (12), for modulating the
5 variable length codes on top of the ADPCM coding, in accordance with the
6 probability of occurrence of data bits en route to the random access memory (12);
7 and
8 a variable-length decoder (15), which is installed in between the random
9 access memory (12) and ADPCM decoder (13), for decoding the variable length
10 codes retrieved from the random access memory (12) to the original ADPCM
11 codes, and then passing on to the ADPCM decoder (13) for restoring to the
12 original audio signals.

13 Applying this model to the example of an electronic anti-shock system
14 (EASS) embedded in ADPCM-based CD players, the variable length encoder
15 (14) is installed in between the ADPCM encoder (11) and the random access
16 memory (12) to further compress ADPCM codes before being saved in the
17 memory, whereby the data storage capacity of the memory can be increased and
18 the buffering time can be extended. Conversely, when the compressed data are
19 retrieved from the random access memory (12), the compressed data are first
20 converted back to ADPCM codes and then to the original audio signals for
21 lossless audio performance.

22 The detailed operation of the variable length encoder (11) and decoder
23 (13) are to be explained in the following paragraphs.

24 The variable length encoder is based on the Huffman coding scheme,

1 whereby pulse codes with the same length bit strings can be converted to variable
2 length codes according to the probability of occurrence of data bits. A variable
3 length code is designed to correspond to the original ADPCM code, but the size
4 of the compressed code is substantially less than a pure ADPCM code. In the
5 Huffman coding scheme, the bit string with the highest occurrence rate will be
6 assigned the variable length code with the shortest length, whereas the bit string
7 with the lowest occurrence rate be assigned the variable length code with the
8 longest length. An example of ADPCM coding is given below using the 3-bit
9 ADPCM mode.

10 In the 3-bit coding mode, the patterns of bit strings are defined as:

11 001 010 011 100 101 110 111

12 The probability of occurrence in a data block are in an ascending order as
13 follows 111, 001, 110, 010, 101, 011, and 100, where the occurrence of 111 has
14 a 0.35 probability; 001 and 110 have a 0.3 probability; 010 and 101 have a 0.2
15 probability; 011 and 100 has a 0.15 probability. Basing on the list of probability
16 of occurrence, the bit string with the highest occurrence rate is assigned a
17 variable length code represented by one data bit "0" to replace the original
18 ADPCM code; the bit string with the second highest occurrence rate is then
19 assigned a variable length code formed by three data bits "10S" (where S being
20 the most significant bit); and the third and fourth highest occurrence rates are
21 respectively assigned three bits codes of 110S and 111S respectively. The
22 conversion table for all variable length codes is shown in Fig. 3. To demonstrate
23 the effect of such a conversion process, a block of data is selected for comparing
24 compressed data before and after the variable length coding process.

1 Assuming the original ADPCM codes are represented by:

2 011_010_001_111_001_110_111_111_001_110_001_110_111

3 The above data block occupies 39 bits, and then the variable length codes
4 are generated on top of the ADPCM coding in accordance with the present
5 invention, the contents of the data block being:

6 1110_1100_100_0_100_101_0_0_100_101_100_101_0

7 and the output data becomes

8 111011001000100101001001011001010, totaling 33 bits on top of the
9 ADPCM coding process, reduced by 6 bits as compared with conventional
10 ADPCM, that means the compression rate is increased by 15%, $(39-33)/39=0.15$.

11 Applying the variable length coding to the above example of EASS,
12 with the increased compression rate, once the data storage capacity of the random
13 access memory is enhanced, the buffering time can be extended. When the
14 compressed data are retrieved from the random access memory, the compressed
15 data are first decoded to the original ADPCM codes and then to the original audio
16 signals for lossless audio output.

17 From the foregoing, the present invention mainly relies on the prognostic
18 code converter to generate variable length codes according to the probability of
19 occurrence of data bits, such that the bit string with the highest probability of
20 occurrence will become the variable length code with the shortest length, and the
21 bit string with lesser probability of occurrence will become the variable length
22 code with longer length. Using this coding technique, the average number of bits
23 in a data block can be substantially reduced and the compression rate can be
24 enhanced accordingly. In the case of EASS, the data storage capacity of random

1 access memory can be increased for extending the buffering time. Therefore, the
2 output performance of the CD player can be improved.

3 The foregoing description of the preferred embodiments of the present
4 invention is intended to be illustrative only and, under no circumstances, should
5 the scope of the present invention be so restricted.